P-26

NASA CASE NO. ARC 12013-1CU

PRINT FIG 1

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Filing Date: 10/4/93
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NASA/ARC

(NASA-Case-ARC-12013-1-CU) MULTI-CHANNEL SPATIALIZATION SYSTEMS FOR AUDIO SIGNALS Patent Application (NASA. Ames Research Center) 26 p

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Unclas

G3/32 0003668

MULTI-CHANNEL SPATIALIZATION SYSTEMS FOR AUDIO SIGNALS

ARC-12013-1CU

AWARDS DIGEST

The invention is directed to generating synthetic head related transfer functions (HRTFs) for imposing reprogrammable spatial cues to a plurality of audio input signals received simultaneously as illustrated in Figure 1 by the use of interchangeable programmable read only memories (PROMs) 16_1 ... 16_4 which store both head related transfer function impulse response data and source positional information as shown in Figure 2 for a plurality of desired virtual source locations, one of which (60°L) is shown in Figures 3A and 3B. The analog inputs 10_1 ... 10_4 of the audio signals are filtered in lowpass filters 12_1 ... 12_4 and converted to digital signals in A/D converters 14_1 ... 14_4 from which synthetic head related transfer functions are generated in the form of linear phase finite impulse response filters 32_1, 32_2 (Fig. 2). The outputs of the impulse response filters are subsequently reconverted to analog signals in D/A converters 20_1 ... 20_4, filtered in lowpass smoothing filters 22_1 ... 22_4, mixed in a pair of summing networks 24_1 and 24_2, and fed to a pair of headphones 18. A simplified method for generating the synthetic HRTFs is employed so as to minimize the quantity of data necessary for HRTF generation.

Invention is believed to reside in the method for deriving HRTFs and the interchangeability of positional information for the virtual source locations through the use of suitably programmed PROMs.

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Employer: National Academy of Science (NRC)
Evaluator:

Serial No.: 08/130,948

Filing Date: 10/4/93

MULTI-CHANNEL SPATIALIZATION SYSTEM FOR AUDIO SIGNALS

Origin of the Invention

The invention described herein was made in the performance of work under a NASA contract and is subject to Public Law 96-517 (35 U.S.C. 200 et seq.) The contractor has assigned his rights thereunder to the Government.

Background of the Invention

Field of the Invention

The invention relates generally to the field of three dimensional audio technology and more particularly to the use of head related transfer functions (HRTF) for separating and imposing spatial cues to a plurality of audio signals in order to generate local virtual sources thereof such that each incoming signal is heard at a different location about the head of a listener.

Description of the Prior Art

Three dimensional or simply 3-D audio technology is a generic term associated with a number of new systems that have recently made the transition from the laboratory to the commercial audio world. Many of the terms have been used both commercially and technically to describe this technique, such as, dummy head synthesis, spatial sound processing, etc. All these techniques are related in their desired result of providing a psychoacoustically enhanced auditory display.

Much in the same way that stereophonic and quadraphonic signal processing devices have been introduced in the past as improvements over their immediate predecessors, 3-D audio technology can be considered as the most recent innovation for both mixing consoles and reverberation devices.

Three dimensional audio technology utilizes the concept of digital filtering based on head related transfer functions (HRTF). The role of the HRTF was first summarized by Jens Blauert in "Spatial Hearing: the psychophysics of human sound localization" MIT Press, Cambridge, 1983. This publication noted that the pinnae of the human ears are shaped to provide a transfer function for received audio signals and thus have a characteristic frequency and phase response for a given angle of incidence of a source to a listener. This characteristic response is convolved with sound that enters the ear and contributes substantially to our ability to listen spatially.

Accordingly, this spectral modification imposed by an HRTF on an incoming sound has been established as an important cue for auditory spatial perception, along with interaural level and amplitude differences. The HRTF imposes a unique frequency response for a given sound source position outside of the head, which can be measured by recording the impulse response in or at the entrance of the ear canal and then examining its frequency response via Fourier analysis. This binaural impulse response can be digitally implemented in a 3-D audio system by convolving the input signal in the time domain with the impulse response of two HRTFs, one for each ear, using two finite impulse response filters. This concept was taught, for example, in 1990 by D.R. Begault et al in "Technical Aspects of a Demonstration Tape for Three-Dimensional Sound Displays" (TM 102826), NASA - Ames Research Center and also in U.S. Patent 5,173,944, "Head Related Transfer Function Pseudo-Stereophony", D.R. Begault, December 22, 1992.

The primary application of 3-D sound, however, has been made towards the field of entertainment and not towards improving audio communications systems involving intelligibility of multiple streams of speech in a noisy environment. Thus the focus of recent research and development for 3-D audio technology has centered on either commercial music recording, playback and playback enhancement techniques or on utilizing the technology in advanced human-machine interfaces such as computer work stations, aeronautics and virtual reality systems. The following cited literature is typically illustrative of such developments: D. Griesinger, (1989), "Equalization and Spatial Equalization of Dummy Head Recordings or Loudspeaker Reproduction", Journal of Audio Engineering Society, 37 (1-2), 20-29; L.F. Ludwig et al (1990), "Extending the Notion of a Window System To Audio", Computer, 23 (8), 66-72; D.R. Begault et al (1990), "Techniques and Application For Binaural Sound Manipulation in Human-Machine Interfaces" (TM102279), NASA-Ames Research Center; and E.M. Wenzel et al (1990), "A System for Three-Dimensional Acoustic Visualization in a Virtual Environment Work Station", Visualization '90, IEEE Computer Society Press, San Francisco, California (pp. 329-337).

The following patented art is also directed to 3-D audio technology and is worthy of note: U.S. Patent 4,817,149, "Three Dimensional Auditory Display Apparatus And Method Utilizing Enhanced Bionic Emulation Of Human Binaural Sound Localization", Peter H. Meyers, March 28, 1989; U.S. Patent 4,856,064, "Sound Field Control Apparatus", M. Iwamatsu, August 8, 1989; and U.S. Patent

4,774,515, "Attitude Indicator", B. Gehring, September 27, 1988. The systems disclosed in these references simulate virtual source positions for audio inputs either with speakers, e.g. U.S. Patent 4,856,064 or with headphones connected to magnetic tracking devices, e.g. U.S. Patent 4,774,515 such that the virtual position of the auditory source is independent of head movement.

<u>Summary</u>

Accordingly, it is an object of the invention to provide a method and apparatus for producing three dimensional audio signals.

And it is another object of the invention is to provide a method and apparatus for deriving synthetic head related transfer functions for imposing spatial cues to a plurality of audio inputs in order to generate virtual sources thereof.

It is a further object of the invention to provide a method and apparatus for producing three dimensional audio signals which appear to come from separate and discrete positions from about the head of a listener.

It is still yet another object to separate multiple audio signal streams into discrete selectively changeable external spatial locations about the head of a listener.

And still yet a further object of the invention is to reprogrammably distribute simultaneous incoming audio signals at different locations about the head of a listener wearing headphones.

The foregoing and other objects are achieved by generating synthetic head related transfer functions (HRTFs) for imposing reprogrammable spatial cues to a plurality of audio input signals received simultaneously

by the use of interchangeable programmable read only memories (PROMs) which store both head related transfer function impulse response data and source positional information for a plurality of desired virtual source locations. The analog inputs of the audio signals are filtered and converted to digital signals from which synthetic head related transfer functions are generated in the form of linear phase finite impulse response filters. The outputs of the impulse response filters are subsequently reconverted to analog signals, filtered, mixed and fed to a pair of headphones. Another aspect of the invention is employing a simplified method for generating the synthetic HRTFs so as to minimize the quantity of data necessary for HRTF generation.

Brief Description of the Drawings

The following detailed description of the invention will be more readily understood when considered together with the accompanying drawings wherein:

Figure 1 is an electrical block diagram illustrative of the preferred embodiment of the invention;

Figure 2 is an electrical block diagram illustrative of one digital filter shown in Figure 1 for implementing a pair of HRTFs for a desired spatial location;

Figures 3A and 3B are diagrams illustrative of the time delay to the left and right ears of a listener for sound coming from a single source located to the left and in front of the listener;

Figure 4 is a graph illustrative of mean group time delay differences as a function of spatial location around the head of a listener as shown in Figure 1; and

Figures 5A and 5B are a set of characteristic curves illustrative of both measured and synthetically derived HRTF magnitude responses for the left and right ear as a function of frequency.

Detailed Description of the Invention

Referring now to the drawings and more particularly to Figure 1, shown thereat is an electronic block diagram generally illustrative of the preferred embodiment of the invention. As shown, reference numerals 10_1, 10_2, 10_3 and 10_4 represent discrete simultaneous analog audio outputs of a unitary device or a plurality of separate devices capable of receiving four separate audio signals, for example, four different radio communications channel frequencies f_1, f_2, f_3 and f_4. Such apparatus is well known and includes, for example, the operational intercom system (OIS) used for space shuttle launch communications at the NASA Kennedy Space Center. Although radio speech communications is illustrated herein for purposes of illustration, it should be noted that this invention is not meant to be limited thereto, but is applicable to other types of electrical communications systems as well, typical examples being wire and optical communications systems.

Each of the individual analog audio inputs is fed to respective lowpass filters 12_1, 12_2, 12_3, and 12_4 whose outputs are fed to individual analog to digital (A/D) converters 14_1, 14_2, 14_3, and 14_4. Such apparatus is also well known to those skilled in the art.

Conventionally, the cutoff frequency f_c of the lowpass filters is set so that the stopband frequency is at one half or slightly below one half the sampling rate, the Nyquist rate f_cN of the analog to digital converters

14_1 ... 14_4. Typically, the filter is designed so that the passband is as close to $f_C N$ as possible. In the present invention, however, another stopband frequency $f_C J$ is utilized and is shown in Figures 5A and 5B. $F_C J$ is specifically chosen to be much lower than $f_C N$. Further, $f_C J$ is set to the maximum usable frequency for speech communication and is therefore set at 10kHz, although it can be set as low as 4kHz depending upon the maximum frequency obtainable from audio signal devices 10_1, 10_2, 10_3 and 10_4.

In Figure 1, the lowpass filters 12_1, 12_2, 12_3 and 12_4 have a passband up to $f_C J$ and include a stopband attenuation of at least 60dB at 16kHz. It should be noted, however, that the closer the $f_C J$ is to 16kHz, the more expensive the filter implementation becomes and thus cost considerations may influence the design considerations. In no case, however, is $f_C J$ chosen to be below 3.5kHz.

Reference numerals 16_1, 16_2, 16_3 and 16_4 denote four discrete digital filters for generating pairs of synthetic head related transfer functions (HRTF), for the left and right ear from the respective outputs of the A/D converter 14_1 ... 14_4. The details of one of the filters, 16_1, is shown in Figure 2 and will be referred to subsequently. Each filtering operation implemented by the four filters 16_1 ... 16_4 is designed to impart differing spatial auditory cues to each radio communication channel output, four of which are shown in Figure 1. As shown, the cues are related to head related transfer functions measured at 0° elevation and at 60° left, 150° left, 150° right and 60° right for the audio signals received, for example, on radio carrier frequencies f_1, f_2, f_3, and f_4.

Outputted from each of the digital filters 16_1 ... 16_4 are two synthetic digital outputs $HRTF_L$ and $HRTF_R$ for left and right ears, respectively, which are fed to two channel digital to analog converters 20_1, 20_2, 20_3 and 20_4. The outputs of each of the D/A converters is then coupled to respective low-pass smoothing filters 22_1, 22_2, 22_3, 22_4. The cut-off frequencies of the smoothing filters 22_1 ... 22_4 can be set to either f_cJ or f_cN, depending upon the type of devices which are selected for use.

The pair of outputs from each of the filters 22_1 ... 22_4 are next fed to left and right channel summing networks 24_1 and 24_2 which typically consist of a well known circuit including electrical attenuations and summing points, not shown. The left and right channel outputs of the filters 22_1 ... 22_4 are summed and scaled to provide a sound signal level below that which provides distortion.

The summed left and right channel outputs from the networks 24_1 and 24_2 are next fed to a stereo headphone amplifier 26, the output of which is coupled to a pair of headphones 18. The user or listener 28 listening over the stereo headphones 18 connected to the amplifier 26 is caused to have a separate percept of the audio signals received, for example, but not limited to, by the four radio channels, as shown in Figure 1, so that they seem to be coming from different spatial locations about the head, namely at or near left 60°, left 150°, right 150° and right 60° and at 0° elevation. Referring now to

Figure 2, shown thereat are the details of one of the digital filters, i.e. filter 16_1 shown in Figure 1. This circuit element is used to generate a virtual sound source at 60° left as shown in Figures 3A and 3B. The digital filter 16_1 thus receives the single digital input from the A/D converter 14_1 where it is split into two channels, left and right, where individual left and right ear synthetic HRTFs are generated and coupled to the digital to analog converter 20_1. Each synthetic HRTF, moreover, is comprised of two parts, a time delay and an impulse response that give rise to a particular spatial location percept. Each HRTF has a unique configuration such that a different spatial image for each channel frequency $f_1 \ldots f_4$ results at a predetermined different position relative to the listener 28 when wearing the pair of headphones as shown in Figure 1.

It is important to note that both interaural time delay and interaural magnitude of the audio signals function as primary perceptual cues to the location of sounds in space, when convolved, for example, with monaural speech or audio signal sound sources. Accordingly, the digital filter 16_1 as well as the other digital filters 16_2, 16_3 and 16_4 are comprised of digital signal processing chips, e.g. Motorola type 56001 DSPs that access interchangeable PROMs, such as type 27C64-150 EPROMs manufactured by National Semiconductor Corp. The PROMs are programmed with two types of information: (a) time delay difference information regarding the difference in time delays TD_L and TD_R for sound to reach the left and right ears for a desired spatial position as depicted by reference numerals 30_1 and 30_2, and (b) sets of filter coefficients used to implement finite impulse response (FIR) filtering, as depicted by reference numerals 32_1 and 32_2, over a predetermined audio frequency range to provide suitable frequency magnitude shaping for left and right channel synthetic HRTF outputs.

The time delays for each channel TD_L and TD_R to the left ear and right ear, respectively, are based on the sinewave path lengths from the simulated sound source at left 60° to the left and right ears as shown in Figures 3A and 3B. A working value for the speed of sound in normal air is 345 meters per second, which can be used to calculate the effect of a spherical modeled head on interaural time differences. The values for TD_L and TD_R are in themselves less relevant than the path length difference between the two values. Rather than using path lengths to a spherically modeled head as a model, it is also possible to use the calculated mean group delay difference between each channel of a measured binaural head related transfer function. The latter is employed in the subject invention, although either technique, i.e. modeling based on a spherical head or derivation from actual measurements, is adequate for implementing a suitable time delay for each virtual sound position. The mean group delay is calculated within the primary region of energy for speech frequencies such as shown in Figure 4 in the region 100Hz-6kHz for azimuths ranging between 0° and 90°. The "mirror image" can be used for rearward azimuths, for example, the value for 30° azimuth can be used for 150° azimuth. The resulting delay actually used is the "far ear" channel while a value of zero is used in the "near ear" channel.

Accordingly, when $TD_L < TD_R$, as it is for a 60° left virtual source S as shown in Figures 3A and 3B, a value for the mean time delay difference in block 30_1 for the left ear is set at zero, while for the right ear, the mean time delay difference for a delay equivalent to the difference between TD_R and TD_L, is set in block 30_2 according to values shown in Figure 4.

For the other filters 16_2, 16_3 and 16_4 which are used to generate percepts of 150° left, 150° right, and 60° right, the same procedure is followed.

With respect to finite impulse response filters 32_1 and 32_2 for the 60° left spatial position, each filter is implemented from a set of coefficients obtained from synthetically generated magnitude response curves derived from previously developed HRTF curves made from actual measurements taken for the same location. A typical example involves the filter 16_1 shown in Figure 2, for a virtual source position of 60° left. This involves selecting a predetermined number of points, typically 65, to represent the frequency magnitude response between 0 and 16kHz of curve 36_1 and 36_2, with curves 34_1 and 34_2 as shown in Figures 5A and 5B.

The same method is used to derive the synthetic HRTF measurements of the other filter 16_2, 16_3 and 16_4 in Figure 1. To obtain the 60° right spatial position required for digital filters 16_4, for example, the left and right magnitude responses for 60° left as shown in Figures 5A and 5B are merely interchanged. To obtain the 150° right position for filter 16_3, the left and right magnitude responses for 150° left are interchanged. It should also be noted that the measured HRTF response curves 36_1 and 36_2 are utilized for illustrative purposes only inasmuch as any measured HRTF can be used, when desired.

The upper limit of the number of coefficients selected for creating a synthetic HRTF is arbitrary; however, the number actually used is dependent upon the upper boundary of the selected DSP's capacity to perform all of the functions necessary in real time. In the subject invention, the number of coefficients selected is dictated by the selection of an interchangeable PROM accessed by a Motorola 56001 DSP operating with a clock frequency of 27mHz. It should be noted that each of the other digital filters 16_2, 16_3 and 16_4 also include the same DSP - removable PROM chip combinations respectively programmed with individual interaural time delay and magnitude response data in the form of coefficients for the left and right ears, depending upon the spatial position or percept desired, which in this case is 150° left, 150° right and 60° right as shown in Figure 1. Other positions other than left and right 60° and 150° azimuth, 0° elevation may be desirable. These can be determined through psychoacoustic evaluations for optimizing speech intelligibility, such as taught in D.R. Begault (1993), "Call sign intelligibility improvement using a spatial auditory display" (Technical Memorandum No. 104014), NASA Ames Research Center.

Too few coefficients, e.g. less than 50, result in providing linear phase FIR filters which are unacceptably divergent from originally measured head related transfer functions shown, for example, by the curves 36_1 and 36_2

in Figures 5A and 5B. It is only necessary that the synthetic magnitude response curves 34_1 and 34_2 closely match those of the corresponding measured head related transfer functions up to 16kHz, which is to be noted includes within the usable frequency range between 0Hz and f_cJ (10kHz). With each digital filter 16_1, 16_2, 16_3 and 16_4 being comprised of removable PROMs selectively programmed to store both time delay difference data and finite impulse response filter data, this permits changing of the spatial position for each audio signal by unplugging a particular interchangeable PROM and replacing it with another PROM suitably programmed. This has the advantage over known prior art systems where filtering coefficients and/or delays are obtained from a host computer which is an impractical consideration for many applications, e.g. multiple channel radio communications having different carrier frequencies f_1 ... f_n. Considering now the method for deriving a synthetic HRTF in accordance with this invention, for example, the curve 34_1, from an arbitrary measured HRTF curve 36_1, it comprises several steps. First of all, it is necessary to derive the synthetic HRTF so that the number of coefficients is reduced to fit the real time capacity of the DSP chip - PROM combination selected for digital filtering. In addition, the synthetic filter must have a linear phase in order to allow a predictable and constant time shift vs. frequency.

The following procedure demonstrates a preferred method for deriving a synthetic HRTF. First, the measured HRTFs for each ear and each position are first stored within a computer as separate files. Next, a 1024 point Fast Fourier Transform is performed on each file, resulting in an analysis of the magnitude of the HRTFs.

Following this, a weighting value is supplied for each frequency and magnitude derived from the Fast Fourier Transform. The attached Appendix, which forms a part of this specification, provides a typical example of the weights and magnitudes for 65 discrete frequencies. The general scheme is to distribute three weight values across the analyzed frequency range, namely a maximum value of 1000 for frequencies greater than 0 and up to 2250Hz, an intermediate value of approximately one fifth the maximum value or 200 for frequencies between 2250 and 16,000Hz, and a minimum value of 1 for frequencies above 16,000Hz. It will be obvious to one skilled in the art of digital signal processing that the intermediate value weights could be limited to as low as f_CJ and that other variable weighting schemes could be utilized to achieve the same purpose of placing the maximal deviation in an area above f_CJ.

Finally, the values of the table shown, for example, in the Appendix are supplied to a well known Parks-McClelland FIR linear phase filter design algorithm. Such an algorithm is disclosed in J.H. McClellend et al (1979) "FIR Linear Phase Filter Design Program", <u>Programs For Digital Signal Processing</u>, (pp.5.1-1 - 5.1-13), New York: IEEE Press and is readily available in several filter design software packages and permits a setting for the number of coefficients used to design a filter having a linear phase response. A Remez exchange program included therein is also utilized to further modify the algorithm such that the supplied weights in the weight column determine the distribution across frequency of the filter error ripple.

The filter design algorithm meets the specification of the columns identified as FREQ, and MAG(dB) most accurately where the weights are the highest. The scheme of the weights given in the weighting step noted above reflects a technique whereby the resulting error is placed above f_CJ, the highest usable frequency of the input, more specifically, the error is placed above the "hard limit" of 16kHz. The region between f_CJ and 15.5kHz permits a practical lowpass filter implementation, i.e. an adequate frequency range between the pass band and stop band for the roll offs of the filters 16_1... 16_4 shown in Figure 1.

Synthetic filters have been designed using the above outlined method and have been compared in a psychoacoustic investigation of multiple subjects who localize speech filtered using such filters and with measured HRTF filters. The results indicated that localization judgments obtained for measured and synthetic HRTFs were found to be substantially identical and reversing channels to obtain, for instance, 60° right and 60° left as described above made no substantial perceptual difference. This has been documented by D.R. Begault in "Perceptual similarity of measured and synthetic HRTF filtered speech stimuli, Journal of the Acoustical Society of America, (1992), 92(4), 2334.

The interchangeability of virtual source positional information through the use of interchangeable programmable read only memories (PROMs) obviates the need for a host computer which is normally required in a 3-D auditory display including a random access memory (RAM) which is down-loaded from a disk memory.

Accordingly, thus what has been shown and described is a system of digital filters implemented using selectively interchangeable PROM - DSP chip combinations which generate synthetic head related transfer functions that impose natural cues to spatial hearing on the incoming signals, with a different set of cues being generated for each incoming signal such that each incoming stream is heard at a different location around the head of a user and more particularly one wearing headphones.

Having thus shown and described what is at present considered to be the preferred embodiment and method of the subject invention, it should be noted that the same has been made by way of illustration and not limitation. Accordingly, all modifications, alterations and changes coming within the spirit and scope of the invention as set forth in the appended claims are herein meant to be included.

APPENDIX - Cont'd.

APPENDIX

SYNTHETIC HRTF MAG. RESPONSE

	FREQ.	MAG(dB)	WEIGHT
1	0	28	1000
2	250	28	1000
3	500	28	1000
4	750	28.3201742	1000
5	1000	30.7059774	1000
6	1250	32.7251318	1000
7	1500	33.7176713	1000
8	1750	34.9074494	1000
9	2000	34.8472803	1000
10	2250	42.8024473	200
11	2500	45.6278461	200
12	2750	42.0153019	200
13	3000	43.1754388	200
14	3250	44.1976273	200
15	3500	42.2178506	200
16	3750	39.4497855	200
17	4000	33.7393717	200
18	4250	33.7370408	200
19	4500	33.3943621	200

APPENDIX - Cont'd.

20	4750	33.5929666	200
21	5000	30.5321917	200
22	5250	31.8595491	200
23	5500	30.2365342	200
24	5750	26.4510162	200
25	6000	23.6724967	200
26	6250	25.7711753	200
27	6500	26.7506029	200
28	6750	26.7214031	200
29	7000	25.7476349	200
30	7250	25.8149831	200
31	7500	27.7421324	200
32	7750	28.3414934	200
33	8000	27.4999637	200
34	8250	26.0463004	200
35	8500	20.0270081	200
36	8750	17.917685	200
37	9000	-3.8442713	200
38	9250	10.077903	200
39	9500	16.4291175	200
40	9750	16.478697	200
41	10000	15.5998639	200

APPENDIX - Cont'd.

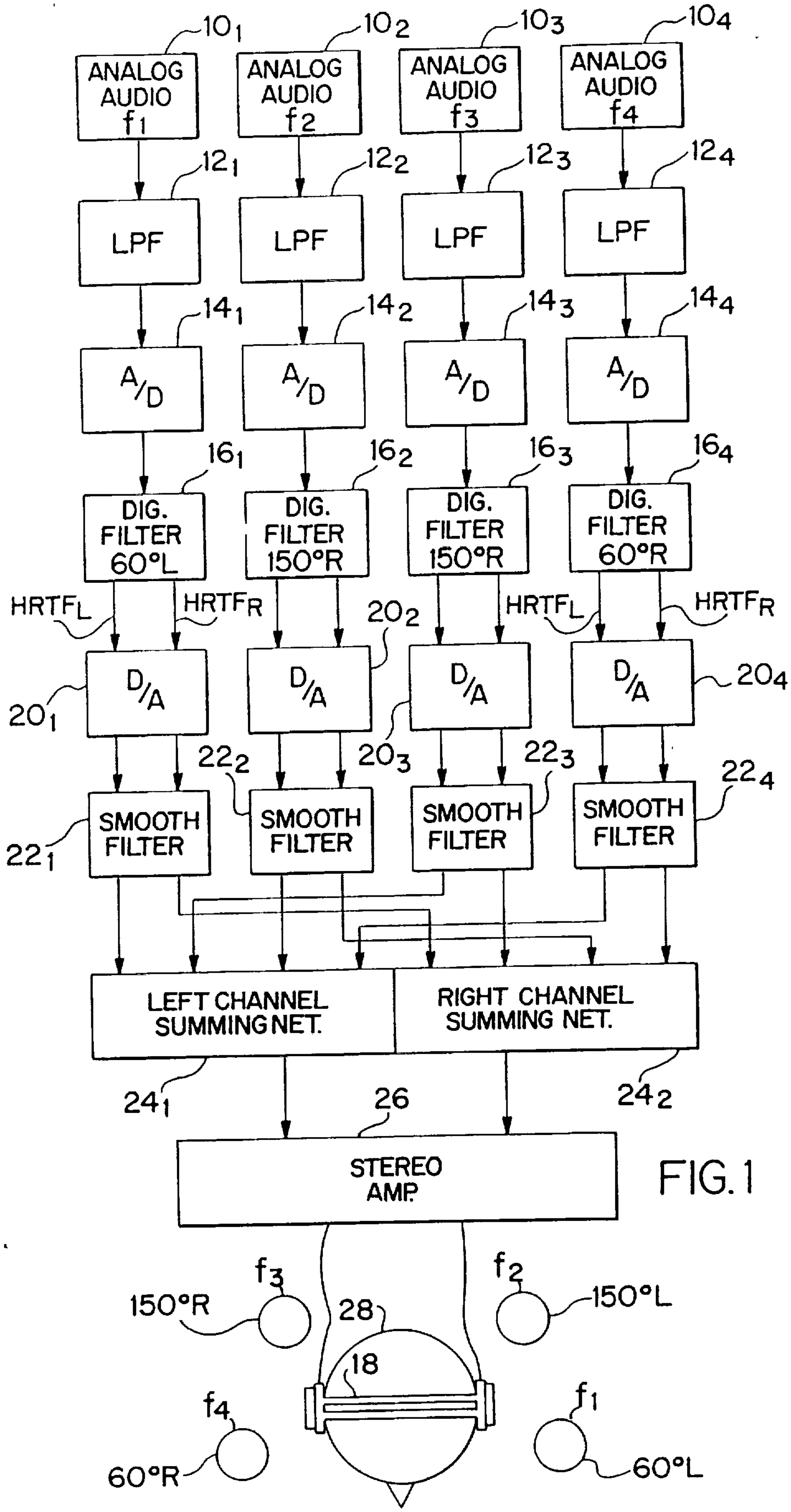
42	10250	13.7440975	200
43	10500	10.9263854	200
44	10750	9.65579861	200
45	11000	6.94840601	200
46	11250	6.51277426	200
47	11500	5.00407516	200
48	11750	6.98594207	200
49	12000	8.66779983	200
50	12250	8.51948656	200
51	12500	6.05561633	200
52	12750	3.43263396	200
53	13000	2.03239314	200
54	13250	0.67809805	200
55	13500	-1.0820475	200
56	13750	-2.7066935	200
57	14000	-4.3344864	200
58	14250	-3.8335688	200
59	14500	-0.4265746	200
60	14750	4.19244063	200
61	15000	7.23285772	200
62	15250	10.9713699	200
63	15500	13.8831976	200

APPENDIX - Cont'd.

64	15750	16.8619008	200
65	16000	18.9512811	200
66	17000	0	1
67	20000	0	1
68	25000	0	1

ABSTRACT

Synthetic head related transfer functions (HRTFs) for imposing reprogrammable spatial cues to a plurality of audio input signals included, for example, in multiple narrow-band audio communications signals received simultaneously are generated and stored in interchangeable programmable read only memories (PROMs) which store both head related transfer function impulse response data and source positional information for a plurality of desired virtual source locations. The analog inputs of the audio signals are filtered and converted to digital signals from which synthetic head related transfer functions are generated in the form of linear phase finite impulse response filters. The outputs of the impulse response filters are subsequently reconverted to analog signals, filtered, mixed and fed to a pair of headphones.



10_1
ANALOG AUDIO f1
10_2
ANALOG AUDIO f2
10_3
ANALOG AUDIO f3
10_4
ANALOG AUDIO f4
12_1
LPF
12_2
LPF
12_3
LPF
12_4
LPF
14_1
A/D
14_2
A/D
14_3
A/D
14_4
A/D
16_1
DIG. FILTER 60°L
16_2
DIG. FILTER 150°R
16_3
DIG. FILTER 150°R
16_4
DIG. FILTER 60°R
HRTF_L
HRTF_R
HRTF_L
HRTF_R
20_1
D/A
20_2
D/A
20_3
D/A
20_4
D/A
22_1
SMOOTH FILTER
22_2
SMOOTH FILTER
22_3
SMOOTH FILTER
22_4
SMOOTH FILTER
LEFT CHANNEL SUMMING NET.
RIGHT CHANNEL SUMMING NET.
24_1
24_2
26
STEREO AMP.
f3
150°R
f2
150°L
28
18
f4
60°R
f1
60°L

FIG. 1

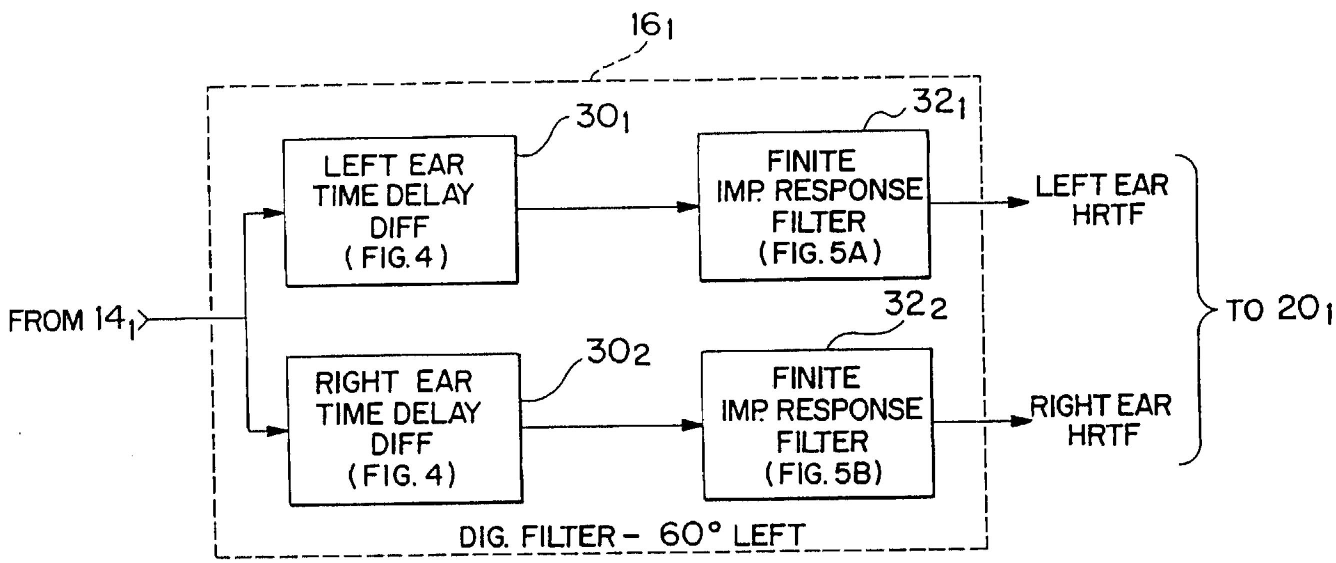


FIG. 2

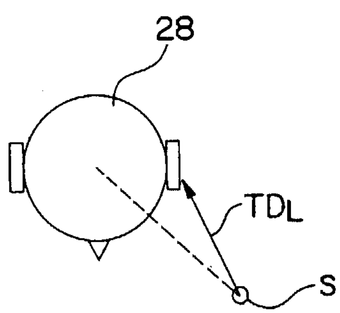


FIG. 3A

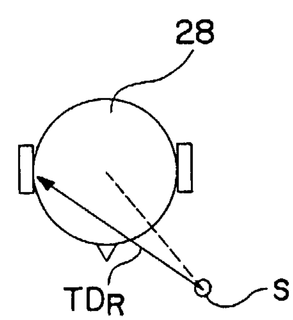


FIG. 3B

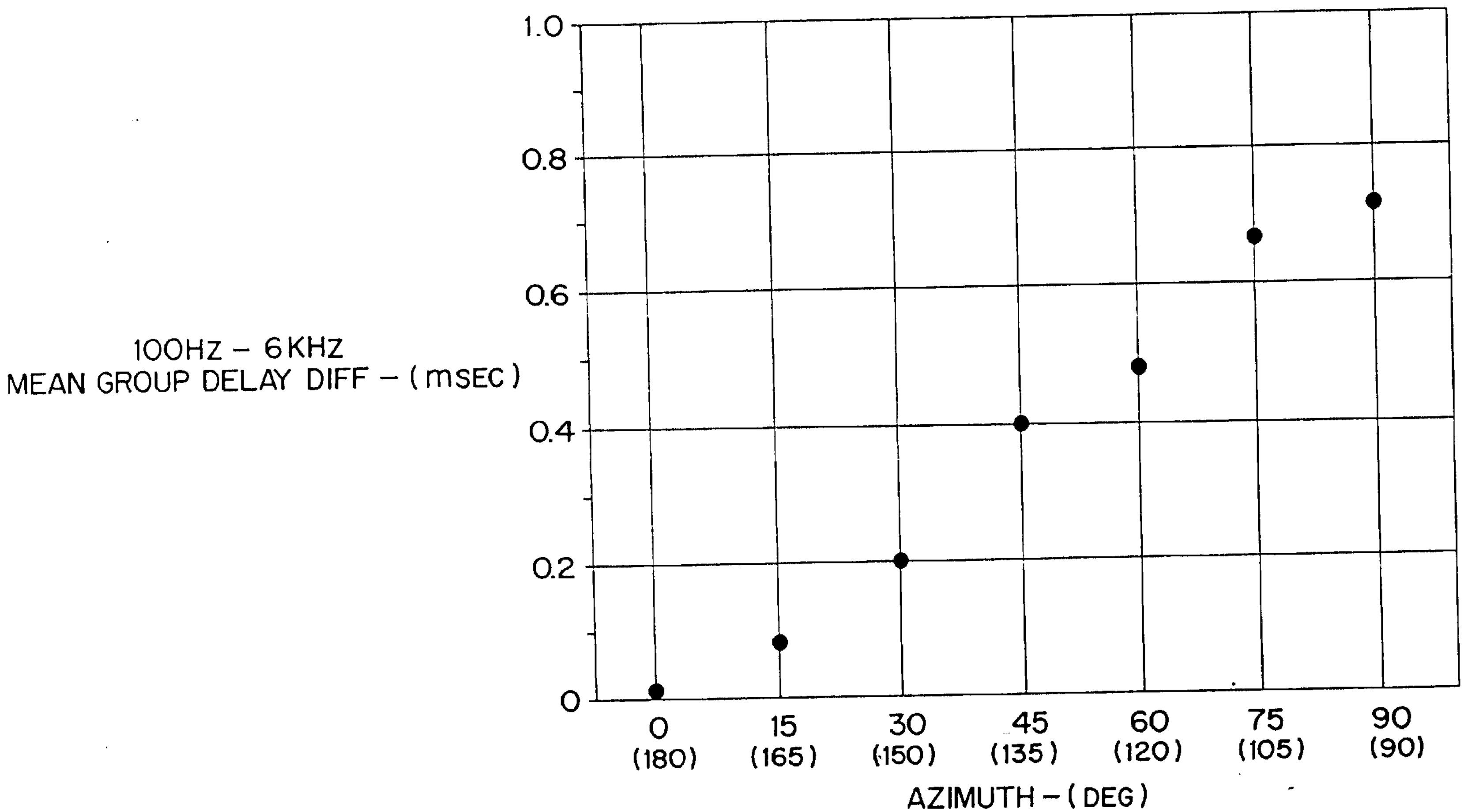

100HZ – 6KHZ
MEAN GROUP DELAY DIFF – (mSEC)
1.0
0.8
0.6
0.4
0.2
0
0
(180)
15
(165)
30
(150)
45
(135)
60
(120)
75
(105)
90
(90)
AZIMUTH – (DEG)

FIG. 4

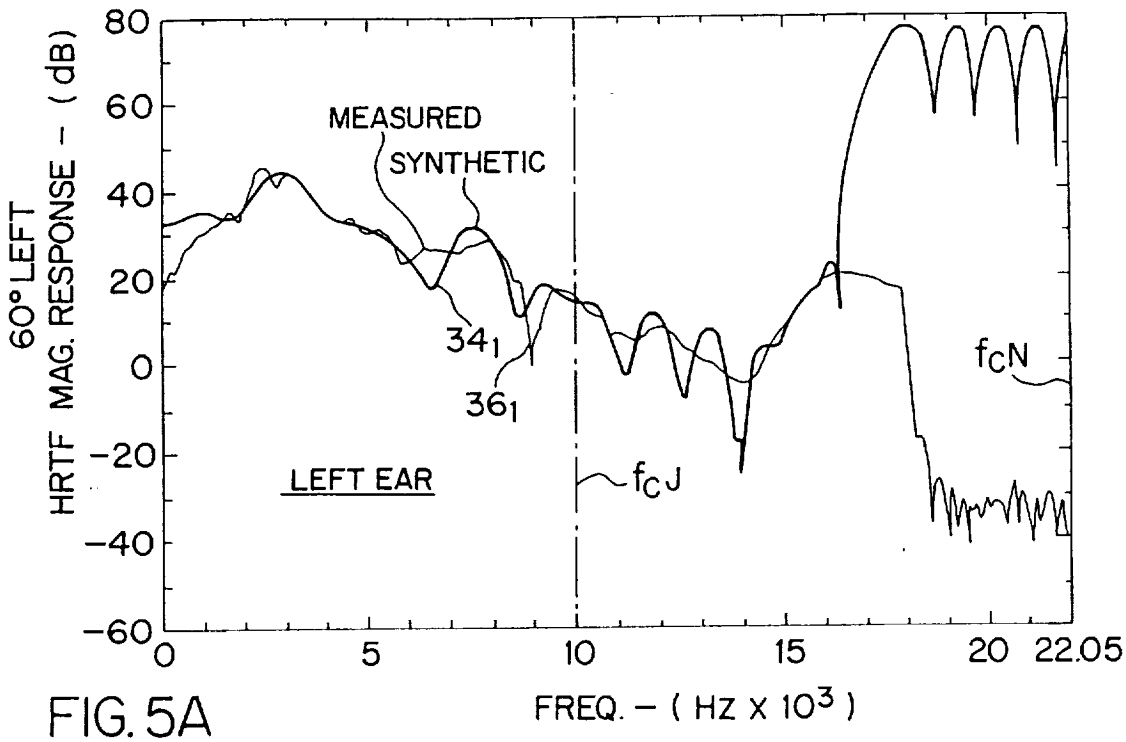

80
60
40
20
0
-20
-40
-60
60° LEFT
HRTF MAG. RESPONSE - (dB)
MEASURED
SYNTHETIC
34_1
36_1
LEFT EAR
f_CJ
f_CN
0
5
10
15
20
22.05
FREQ. - (HZ X 10^3)

FIG. 5A

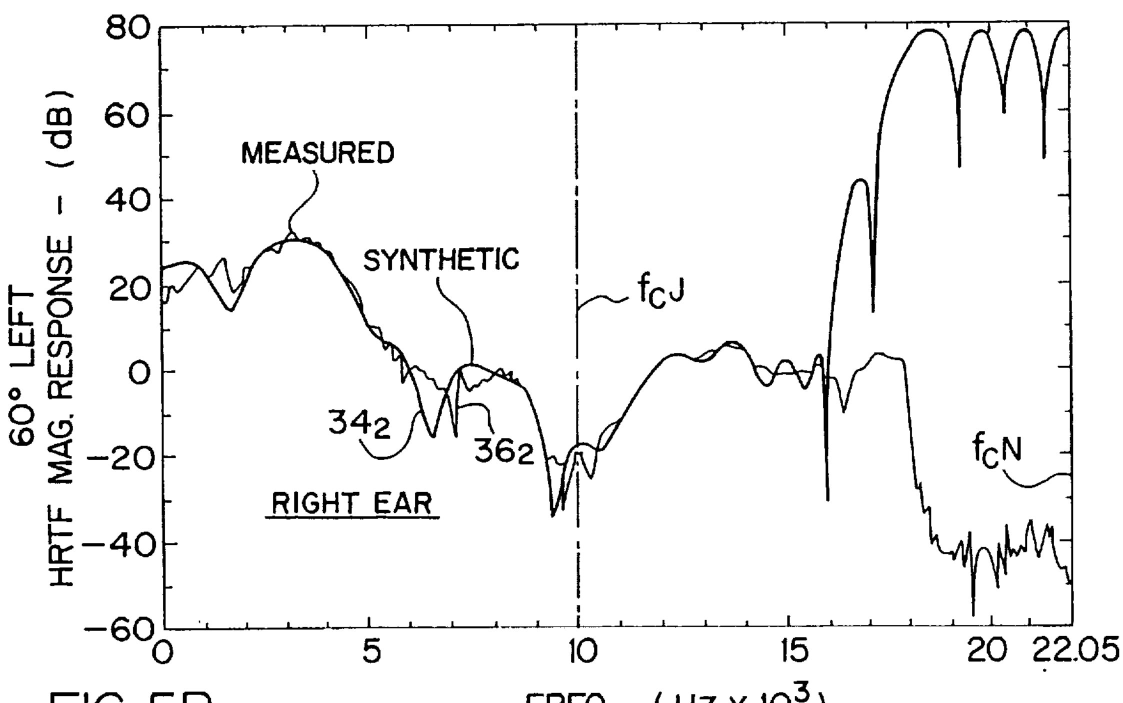

80
60
40
20
0
-20
-40
-60
60° LEFT
HRTF MAG. RESPONSE - (dB)
MEASURED
SYNTHETIC
34_2
36_2
RIGHT EAR
f_CJ
f_CN
0
5
10
15
20
22.05
FREQ. - (HZ X 10^3)

FIG. 5B